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Development of Protocols for Maritime Mobile Communications

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13. ABSTRACT *(Maximum 200 words)*

Data communications are becoming more extensively used in the maritime mobile services. With no current general protocol profile that can be used for data communications between mobile services, it will be advantageous to develop protocols that adhere to the Open System Interconnection (OSI) standards. Implementation of such protocols will allow multiple shipboard equipment to communicate via a shipborne network and then transmit the data to a shore-based network in an effective and efficient manner. Automatic Repeat reQuest (ARQ) techniques are often used by packet-switching networks to provide error-free communication links between network nodes. Information throughput is highly link dependent; as the noise or interference on the link increases, throughput decreases. To improve the throughput on a packet switching communications network, an adaptive ARQ strategy is developed and applied to the Stop-and-Wait protocol. A comparison of the throughput efficiencies of the simulated adaptive SW protocol with the non-adaptive SW protocol showed a marked improvement in throughput when the communication links are subjected to high channel bit error rates.

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INTRODUCTION

As new systems using computers and computer-based equipment are implemented in the mobile maritime services, ships will have an increased requirement to transmit data to and from multiple computer/computer-based equipment. Although a portion of the required data communications will occur between the equipment aboard ship, responsive end-to-end data communications must be assured between the ship-borne systems and terrestrial data processing centers. The internetworking of the terrestrial and mobile ship systems pose several problems.

While it is recognized that the availability of hosts/routers in the network will vary the topology of the network on a time basis, mobile systems, not being stationary, move from sub-network to sub-network, changing the topology on a geographic basis. As a result, the network must transparently adapt to the changing network topology in both time and geographic distribution to maintain continuous connectivity. The network addressing plan must allow unambiguous identification of any ship-borne host computer on a global basis and must be able to route data packets internationally throughout the changing network topology.

The vast geographical distribution of the network nodes requires that data communications be radio-based. Radio frequency (RF) systems are subjected to channel-limited data rates, multipath fading, interference/jamming, and high noise conditions that limit the throughput efficiency of the network in varying degrees from MF to UHF as compared to constrained paths.

To provide complete end-to-end service, the maritime mobile services must be compatible with terrestrial networks which are converging to the Open Systems Interconnect (OSI) standard. Use of the OSI standards would allow multiple shipboard equipment to communicate via a ship-borne network and then transmit the data to the shore-based network. The shore-based network would then transmit the data to the required destination via specialized networks or by use of the public switched telecommunications networks. Additionally, this implementation would not limit the size, capacity, manufacture or data networking capabilities of the on-board computers or computer-based equipment.

OSI is based on the concept of cooperating distributed applications and is concerned with information exchange between systems rather than the internal operation of the systems. The OSI basic reference model as defined by CCITT Recommendation X.200 and ISO 7498 is based on a seven layer communications hierarchy; each with a distinct function. The upper layers (5 through 7) are concerned with interconnection of applications on the processors. The lower layers (1 through 4) deal with the interconnection of processors and connections through which data in any format can move from source to destination. Each layer within the OSI model performs a logically related set of functions required to communicate with another system. It relies on the next lower layer to perform more primitive functions and to conceal the details of those functions. In the same manner, it provides services to the next higher layer on a transparent basis, that is, it performs a set of services required by the next higher layer without exposing the higher layer to the details of implementation. In this way, changes may be implemented within a layer without affecting the method of implementing other layers. Figure 1 shows the OSI network architecture and a block diagram representing a communications network between n stations.

The primary thrust of this paper is at the data link layer of the OSI model. It is the data link layer that provides for the reliable transfer of data between stations on the communications network. The high noise conditions discussed above will often result in a large number of errors in the received message, requiring a significant increase in the number of retransmissions. At the data link layer, it will be shown that an adaptive automatic repeat request (ARQ) strategy may be employed to enhance the throughput of a network operating in high noise conditions by varying the length of the message by reducing or increasing the information contained in the packet. The increases or decreases in packet length will be made based on the number of consecutive successful transmissions or consecutive retransmissions of a packet. Keeping track of the consecutive packet transmissions and using this information to determine when to adapt the packet length should only require software upgrades of the protocols currently used.

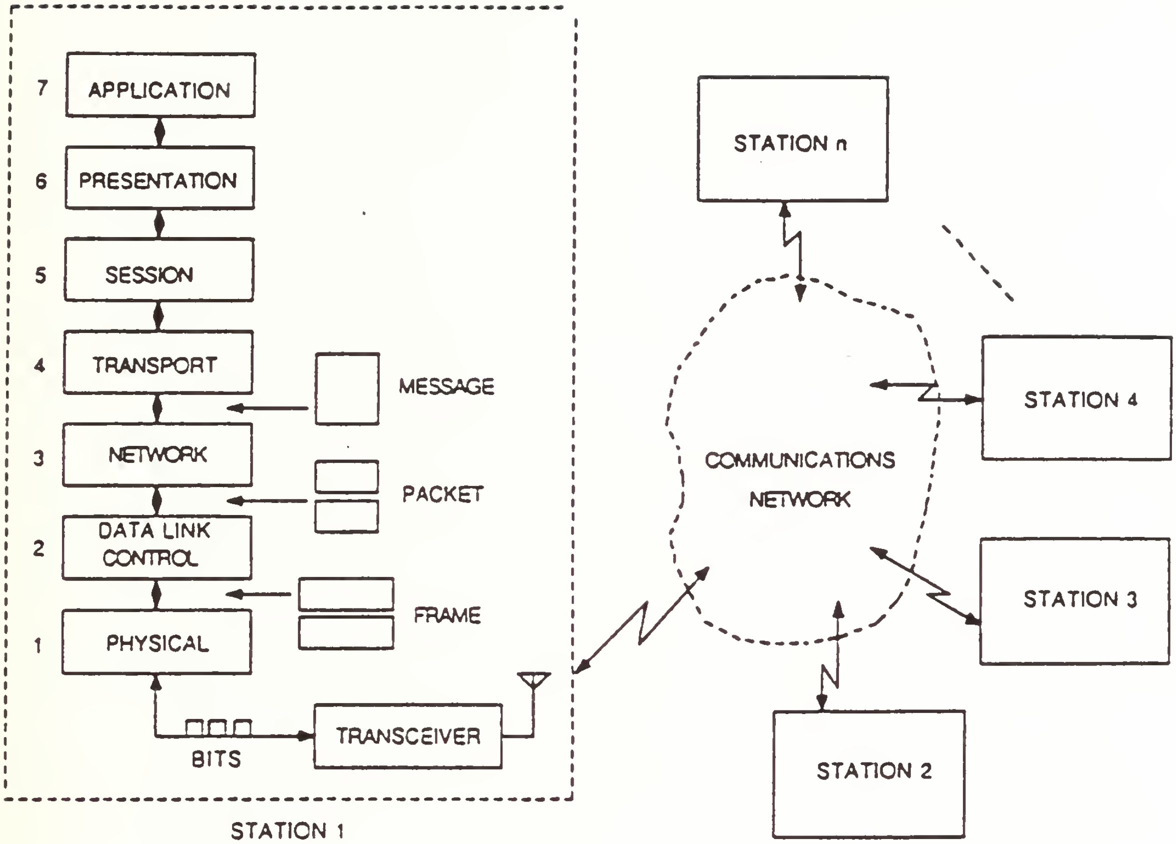

Figure 1 OSI network architecture and data communications network

A secondary thrust of this paper is at the network layer. Within this layer, the implications of mobile nodes within the network on the addressing schemes and routing architectures will be discussed briefly.

DATA LINK LAYER PROTOCOLS

The data link layer (layer 2) and the network layer (layer 3) provide the upper layers with a virtual link over which messages are transferred to other stations on the communications network. The upper layers expect the network layer and data link layer to perform the functions necessary to reliably deliver the message to the receiving station

The network layer works closely with the data link layer to break the message into packets for transmission. Overhead control bits are added to each packet to form a frame. The data link layer establishes a bit sequence to identify the frame starting and ending points This bit sequence must be unique and no inadvertent occurrences of the sequences can be transmitted prior to sending the bit sequence marking the end of the frame The data link layer must account for the frames that have been transmitted and the acknowledgments identifying receipt from the receiving station. Additionally, the data link layer is responsible for ensuring that the frames are received in the proper order so the message may be reassembled and passed in the correct form to the layer above The data link layer is responsible for providing the upper layers with an "error-free" communications link although the physical layer (layer 1) provides an unreliable bit pipe through which to transmit the actual bits. Therefore, the data link layer must check for error-free reception before accepting the each frame Finally, the data link layer must be able to establish and control the communications link between the stations

A protocol is a set of specific procedures for data transfer that the data link layer uses to obtain error-free communications when using the unreliable channel provided by the physical layer. These protocols or sets of procedures must be designed to allow the data link layer to identify and correct all possible transmission errors. One technique used to correct these transmission errors is called automatic repeat request (ARQ). This technique draws on the ability of the data link layer to detect frame errors which occur during transmission and request the sender retransmit the frame To implement these ARQ protocols, the data link layer establishes a specific format for the frames and transfer procedures to handle the data transfer

One type of ARQ protocol is called the Stop-and-Wait (SW) protocol In the SW protocol, a single frame is transmitted and the transmitter stops and waits for a response from the receiving station. If the frame is received without errors, then an acknowledgment (ACK) will be returned from the receiver and the next frame is transmitted. If an error is encountered, then a request for retransmission (NAK) will be returned and the same frame is retransmitted. If no response is received, the sender only waits a certain amount of time referred to as a *time-out* and then the frame is retransmitted. The SW protocol is effective for data communications when the round trip propagation delay time for a packet is short relative to the frame transmission time This protocol becomes inefficient in using the available bandwidth when the round trip propagation time is long or when stopping the transmission introduces additional delays associated with resynchronization of the transmission equipment.

To make better use of the bandwidth on these channels with long propagation delays, continuous or "pipelining" protocols are used. A pipelining protocol allows the transmitter to send frames continuously without stopping and waiting for a response for each frame. There is a limit on how many frames can be sent without having received a response from the receiving station for the first of the unacknowledged frames This limit is referred to as a "sliding window" and is determined by the system capabilities and the propagation delay. For the most efficient operation, the window should be greater when the propagation delay is longer [1]

Go-Back-N (GBN) is a pipelining protocol which allows a transmitter to send up to N-1 frames before stopping and waiting form a response from the receiver The receiver, in contrast to the transmitter, has a receive window equal to one frame The receiver is looking to receive a specific frame and once that frame is received, it immediately looks for the next one Typically, the receiver discards a frame received with errors When a later frame arrives correctly and is out of sequence, or not the frame the receiver was expecting, the transmitter is notified that an error has occurred When the transmitter learns that one of the frames was received with errors, it backs up and begins to transmit again starting with that frame The beginning of the window slides up to the frame being retransmitted, so this frame is the first of N-1 possible frames in the new window. All the frames that were transmitted following the frame received in error are also retransmitted If the protocol is operating in an environment which causes several consecutive packets to incur errors during transmission, a large number of frame retransmissions will result Under these conditions, the transmission delays can be quite large and the protocol efficiency quite low.[1]

Selective Repeat (SR) is another pipelining protocol. Similar to GBN, the SR protocol uses a sliding window to allow N-1 frames to be sent without waiting for a response However, the SR protocol employs a different method of error correction than GBN For the SR protocol, only frames received in error are retransmitted. Since the data link layer must still provide the message in the proper form to the layer above, frames received without error must be stored in a buffer while waiting for the retransmitted frames. This selective repeat feature adds complexity to the system and additional memory requirements.[1]

ADAPTIVE ARQ STRATEGY DEVELOPMENT

GENERAL CONSIDERATIONS

The effectiveness of information transfer of the typical ARQ protocol depends largely on the average channel bit error rate. For a low and generally stable channel bit error rate, the optimum packet size for the maximum throughput efficiency can be determined. Throughput efficiency is defined as the amount of actual information transferred relative to the total number of bits transmitted. Since every packet transmitted has the same amount of overhead required by the ARQ protocol, the larger the information field, the better the packet efficiency. Unfortunately, as the packet size increases the probability of an error occurring during transmission increases correspondingly. Generally, the maximum throughput efficiency is realized by utilizing the optimum packet length for the link conditions and minimizing the number of retransmissions required. For long term changes in the channel bit error rate, the optimum packet lengths can be modified manually to maximize the throughput efficiency. If the channel bit error rate increases quickly, the typical ARQ strategy will continue to retransmit packets which were received in error at the packet length determined to be optimum for the average channel bit error rate. For these typical ARQ strategies the throughput efficiency suffers greatly during the periods of operation with high bit error rate.[1]

The channel bit error rate does not often remain constant for a RF system, particularly for a system affected by signal fading or subject to interference and jamming. Destructive interference of the RF signal as a result of multi-path interference may result in an abrupt, large increase in the channel bit error rate. This condition may only last for several packet transmissions followed by an equally abrupt decrease in the bit error rate. A less destructive degradation of the channel, such as an increase in noise level, may result in a less abrupt and severe increase in the bit error rate. This noisy degradation of the channel may last longer than the quick destructive interference, causing an increase of the bit error rate to persist for a large number of packet transmissions. Since the bit error rate does not remain constant over the channel and may actually vary widely, an ARQ strategy which adapts the packet length for a changing bit error rate would increase the throughput efficiency over one that does not

Different types of signal fading conditions and channel noise dictate that different adaptive strategies be employed. Since many different channels conditions exist, developing a different strategy for each condition is nearly impossible and certainly impractical. Therefore, two strategies with some optional design variations were developed in an attempt to counteract the effects of signal fading and increased channel noise conditions.

ADAPTIVE STRUCTURE

An adaptive ARQ strategy decreases the length of the packets by decreasing the amount of data transferred in the information field. This reduction of data occurs for packets which have already failed a certain number of retransmission attempts, indicating an increase in the bit error rate on the forward channel. Also, the lengths of packets not yet transmitted for the first time may be increased (up to a maximum size allowed) if the previous packets are transmitted without error, indicating that the forward channel bit error rate has decreased.

The structure for the adaptive strategies can be viewed as consisting of various levels, where each level equates to a packet length, as shown in figure 2. The top level of the structure is associated with the largest information field, and is typically the starting point for a communications session. Each level below the top has an information field length associated with it which is less than the largest information field.

Each successive level is connected by a series of steps. The number of steps between different levels does not have to be the same. Once the structure is established, the number of steps are fixed for that structure. These steps connecting the levels determine how many retransmission attempts at a certain length are made prior to the modification of the packet length to the new length associated with the new level. A step up to the next step or level of the structure is made for each successful packet transmission until the top level is reached. For each packet which suffers an error in transmission, a step down is made until the bottom level is reached.

In figure 2, the set of steps between two levels have a packet length equal to the length associated with the upper level. In this figure, to go from level $L(0)$ to $L(1)$ requires four steps. This means that four consecutive attempts are made to transfer a packet with length equal to the maximum length associated with $L(0)$. If four consecutive attempts fail to transfer the packet error-free, then level $L(1)$ is reached, and the packet length is modified prior to the next retransmission attempt.

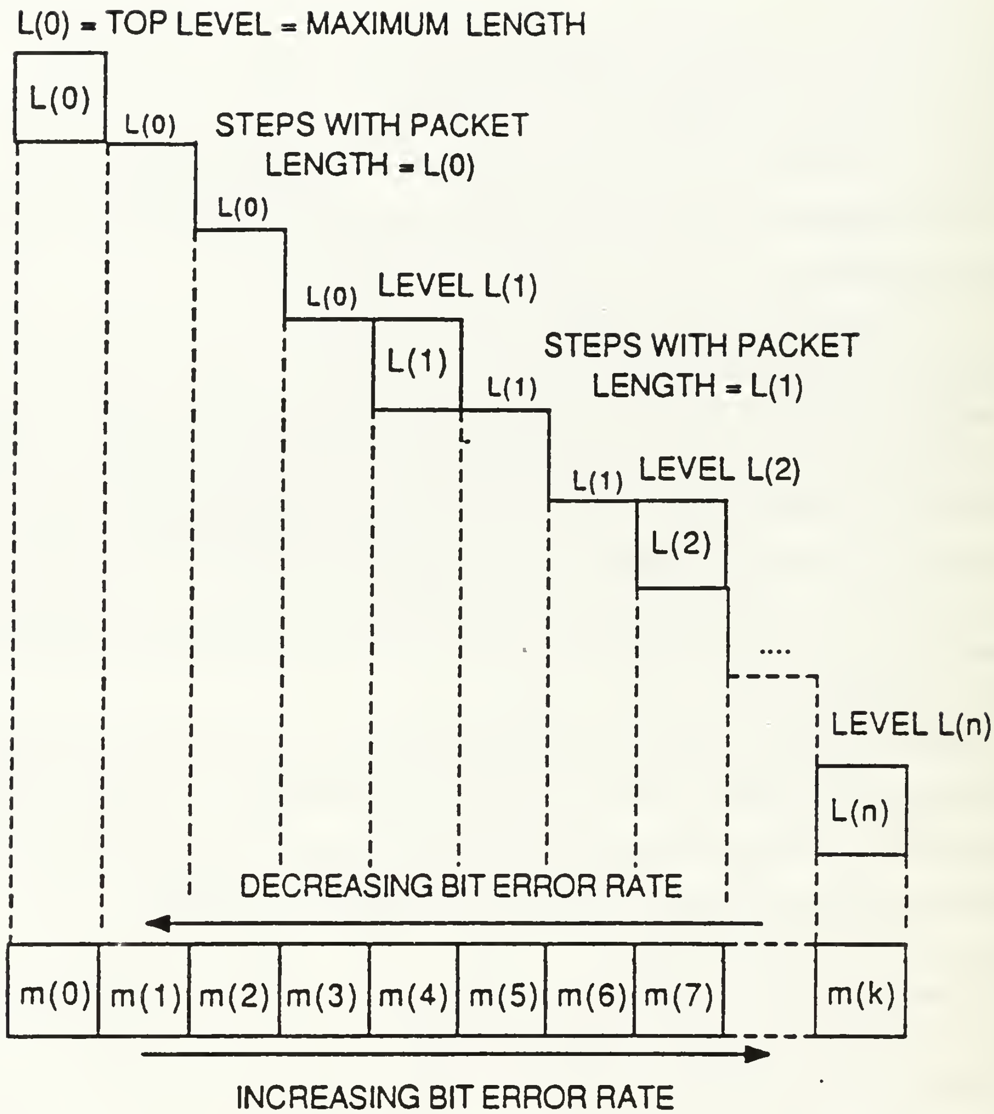

Figure 2 Adaptive strategy structure and state vector

The adaptive structure can be represented by a state vector **m**. Each of the elements or the states of the state vector $m(i)$, $i = 0,1,\ldots,k$, correspond to either one of the levels or the steps of the adaptive structure. The number of states $(k+1)$ in the state vector is equal to the number of levels and steps in the adaptive structure. Moving left or right to a new state in the vector corresponds to stepping up or down the adaptive structure, respectively. The value of each state $m(i)$ is the packet length associated with the corresponding level or step of the structure. Since some levels and steps have the same associated packet length, the values of some states will be the same. For example, as shown in figure 2, when the current state is $m(3)$, then the packet is transmitted at the length associated with level $L(0)$. If the packet is received in error, then the state is updated to $m(4)$ and the length is modified to that associated with $L(1)$. This state vector will be used by the station software to implement the adaptive strategy. The packet length to be transmitted by the station is determined by the value of the current state of the vector. The ends of the state vector correspond to the levels associated with the maximum and minimum packet lengths of the structure. When the state at the left-hand end of the vector is reached, this corresponds to reaching the top of the adaptive structure and the system state will remain at this vector element until a packet suffers an error during transmission. Likewise, when the right-hand end of the vector is reached, the state corresponds to the bottom of the structure and the packet is retransmitted at the minimum length associated with this state until it is successfully transferred.

As an example of one alternative method of designing the steps between levels is shown in figure 3. In this structure, a different set of steps or path is provided when ascending from one level to another. This strategy may be useful in the situations where additional packet transmissions at the length associated with the lower level are desired prior to modifying the packet lengths to the longer length of the level above. The steps ascending to the next level are desired can be variable in number, and may intersect any of the descending steps or go all the way up to the next level. The state vector **m** for this structure is made up of different segments. The states $m(i)$ in each segment of the state vector correspond to one level and the steps which ascend and descend from that level enter the next segment of states, which all have the same associated packet length. As in the previous vector, moving left or right corresponds to stepping up and down the structure. For this vector, the arrows indicate which is the next state when moving from one segment to another. This method of transition between levels provides the system designer with added flexibility to optimize the system throughput efficiency.

When the signal is subjected to fading or jamming on the forward channel, there is a correspondingly large increase in the bit error rate which occurs quickly relative to the time required for packet transmission. Under these conditions, the packet length must be modified substantially to gain an increase in the information throughput. The type of adaptive strategy to counteract the conditions imposed by this large chaging bit error rate is a two-level adaptive strategy , shown in figure 4.

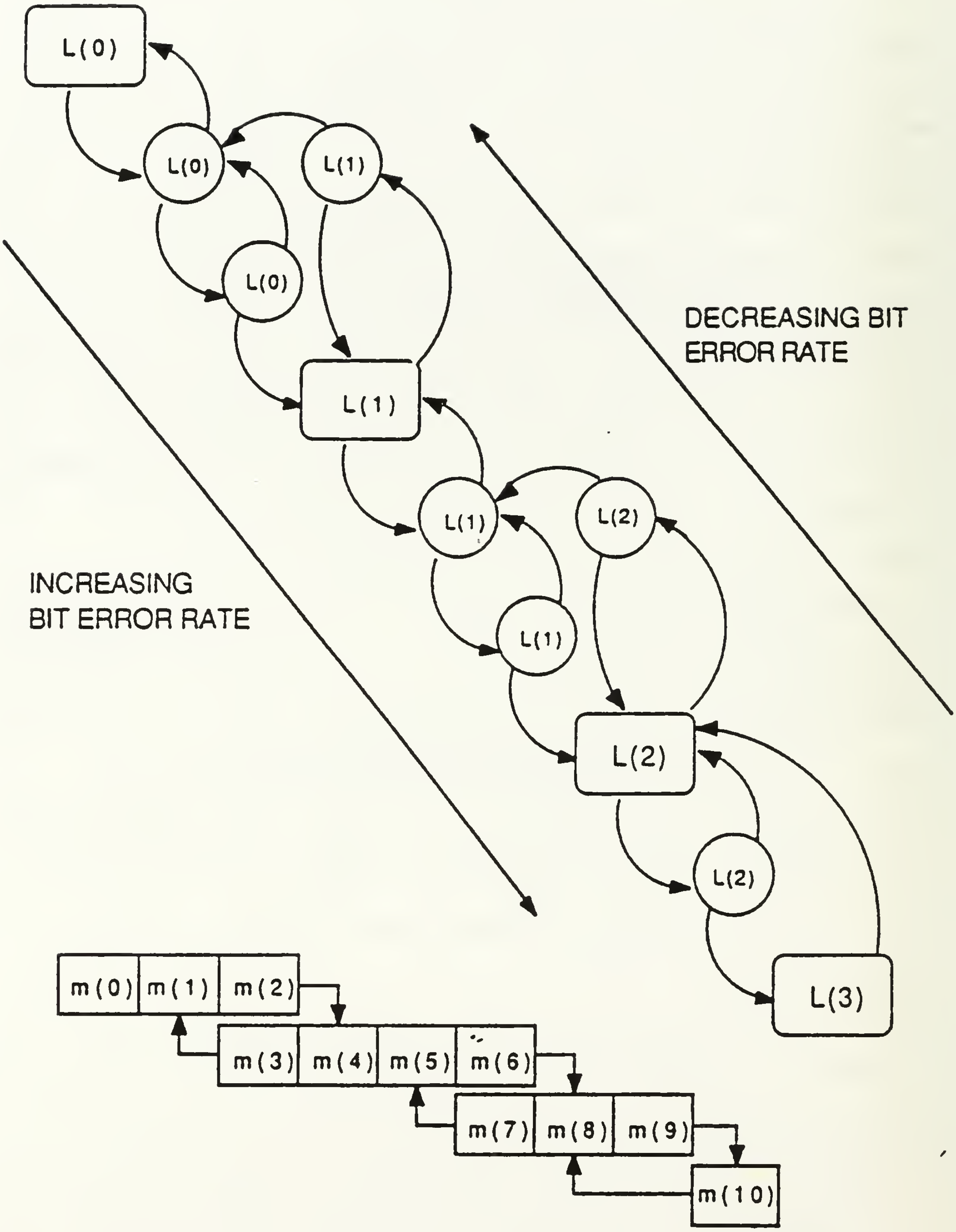

Figure 3 Alternative adaptive strategy structure design

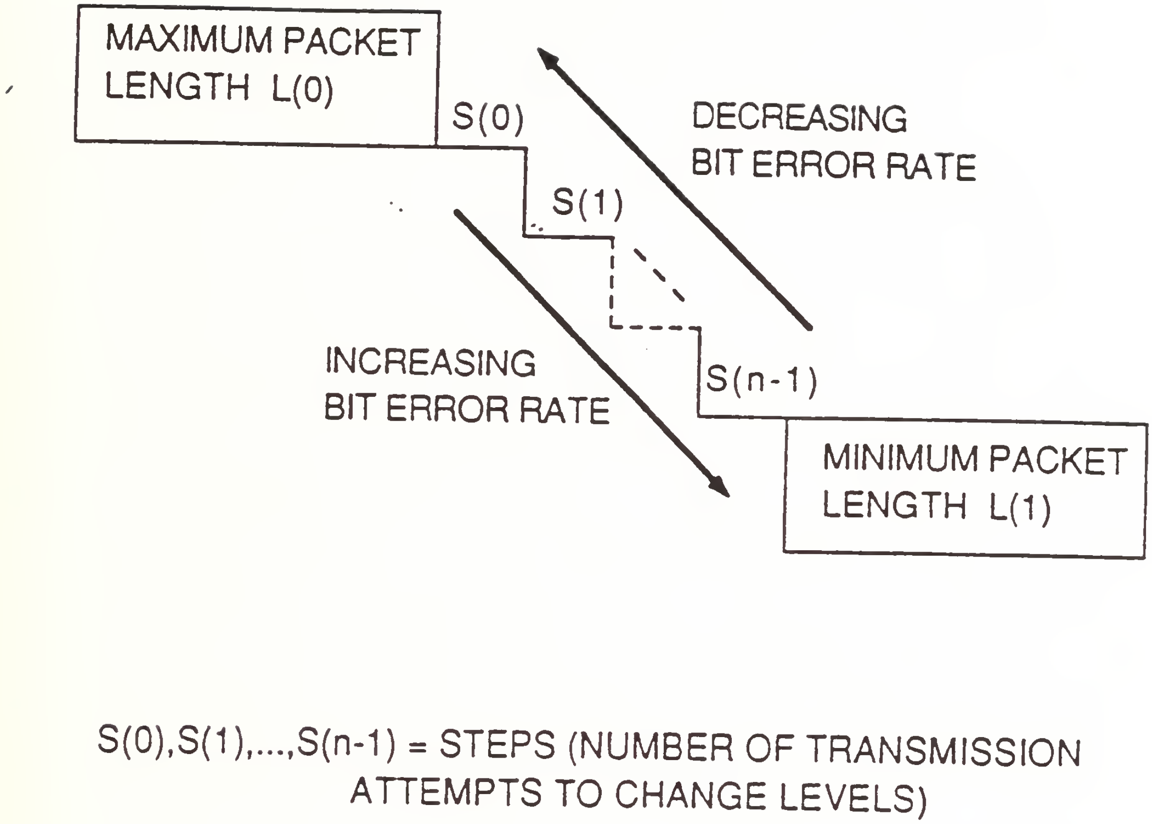

Figure 4 Two-level adaptive strategy

The structure of the two-level strategy, as the name implies, transmits only packets with two lengths. The top level is the maximum packet length, and would typically be the length determined to be optimum for the average channel bit error rate of the system employed The packet length associated with the lower level is the length determined to be optimum for the system during periods of operation under the high bit error rate conditions which occur most frequently. These channel bit error rates are usually not known for a system, and must typically be determined from empirical testing of the system and results accumulated for long periods of time.

The steps connecting the levels which determine the number of retransmission attempts or the number of consecutive, successful packet transmissions prior to a length modification are completely variable. Using the alternative structure design, the number of steps to descend to the lower level can be specified to be different from the number of steps to ascend to the top level. This variability allows a system designer to specify how quickly the system will react to a perceived increase or decrease in the forward channel bit error rate. Determining the number of steps between the levels which optimizes the system throughput efficiency, similar to the level determination, is very system dependent. System testing under as many conditions as possible would be required to find the number of steps which optimize the system throughput.

A different strategy is needed in the cases where varying noise, which may be modeled as Gaussian noise, causes an improvement or degradation of the channel quality, resulting in a forward channel bit error rate which changes accordingly. The changing bit error rate may vary slowly relative to the time required for packet transmission, and in smaller amounts than in the signal fading case. For this type of changing bit error rate condition, the adaptive strategy must vary the packet length for the channel conditions. The type of adaptive strategy shown in figure 2 is multi-level adaptive strategy, and will step or modify the packet length to various lengths associated with each level ranging from the maximum to the minimum packet length allowed.

Similar to the two-level strategy, the top level is associated with the maximum packet length allowed. This packet length is typically chosen to optimize the system operating with the most frequently occurring bit error rate. Because this strategy targets the cases where the changes in the bit error rate are less severe than in the two-level strategy, the information field lengths of each lower level are typically no less than one-half the length of the level above. The number of levels and steps in the multi-level are completely variable. As the channel bit error rate increases, after a specified number of failed retransmission attempts of a packet with length greater than the minimum length allowed, the packet length is changed to the next lower level and retransmitted. This procedure is continued until the packet is successfully transmitted or the lowest level is reached. Once a specified number of packets at a certain level have been successfully transmitted, the packet length of the next packet to be transmitted is increased to the next level. As long as packets are transmitted successfully, stepping up the packet length to the next level continues until the highest level is reached and the packet length is the maximum for the system.

ADAPTIVE STRATEGY IMPLEMENTATION

The main feature of the adaptive strategies is the modification of the length of the message packet after a specified number of consecutive retransmission attempts or successful transmissions. All ARQ protocols require that the transmitting station store all transmitted packets until their successful transfer is acknowledged by the receiving station. Since the transmitted information is stored until acknowledged, it can be broken up into smaller packets to be retransmitted at lengths associated with the new level of the adaptive structure. For the case where the bit error rate decreases and the packet length is to be increased, the number of information bits in the information field is simply increased accordingly.

The addition of the adaptive feature is designed to be strictly a software modification which only adds some additional overhead requirements to the transmitting station. The transmitting station must know what information has been successfully transferred, the information that is attempting to be transferred, and the beginning of the next block to be transferred. In addition, the number of attempted transfers at the current packet length, as well as the number of consecutive packets transmitted successfully, must be accounted for. The addition of a state vector for the two-level or multi-level strategy can take care of most of the additional packet accounting required. The state vector is used by the transmitting station software to implement the adaptive strategy, and identifies the proper length of the packets to be transmitted at any particular time. The adaptive strategy can be designed to be as complex or as simple as the system designer determines necessary to optimize the system throughput.

The state vector is always updated when an acknowledgment for receipt or rejection of a transmitted packet is received. For an accepted packet, the state is updated by moving one position to the left in the vector, and the next packet of information is assembled into a frame at the length specified by this new state and transmitted. If a frame is rejected, the state is updated by moving to the right, and the new state specifies if the packet length is to be modified. When modification of the packet length is indicated prior to retransmitting the frame, the local station must verify that the receive sequence number $N(r)$ of the remote station is the same as the send sequence number $N(s)$ of the frame that the local station is attempting to transfer. If the received frame from the remote station which contained the negative acknowledgment (NAK) was received without error, then the verification is complete. But, if the frame which contained the NAK is received with errors or not received at all, the verification procedure can be accomplished through a P/F exchange.[2] A flow diagram of the verification procedure is shown in figure 5. The local station transmits a supervisory command frame with the P bit set to '1', which forces the remote station to send a supervisory response frame with the F bit set to '1'. The remote station also provides its current $N(r)$ value in this response frame. Then, if a modification of the packet length is required, the frame is reassembled with the new packet length, a new cyclic redundancy check (CRC) is performed, and the checksum is placed in the CRC field. The frame is then retransmitted, and the data transfer process continues.

ADAPTIVE SW ARQ

In the following discussion, examples of adaptive SW ARQ are presented. While these do not represent the long distance, long propagation delay conditions experienced mobile maritime communications in general, they do provide examples of short range ship-to-shore communications and provide perspective on how the throughput efficiency may be increased through the use of adaptive strategies.[*] Similar improved results are expected for the GBN and SR ARQ protocols. However, further research is required to verify this.

Point-to-point communications using adaptive SW protocols operate in a very similar manner to typical communications with SW protocols.[4] The sessions are initiated in the same way, and when operating in the asynchronous balanced mode (ABM) the information can flow in both directions.[4]

Figure 6 conceptually illustrates some typical operations for point-to-point communications sessions between two stations A and B. These examples are for short range, RF packet switching networks which may be subject to noise and signal fading, so the probability P_b may be different for the channel from A to B than from B to A. Both stations are combined stations, that is they may act as the primary or secondary station during data transfer. The mode used for the data transfer is ABM. In these examples, information frames are represented by a single 'I' symbol of various lengths for each frame. The maximum length for each frame is specified by the current state $m(i)$ of the state vector $\mathbf{m}$. The supervisory (S) and unnumbered (U) frames [2-4] are specified by the short, double 'II' symbols and are all transmitted at a length of 48 bits. The length of the frame symbols represents the time required to transmit the frame, including the small amount of time used to process the frame. The slanted lines indicate amount of propagation delay in the transmission of the frames. These lines are only slightly angled, which reflects how small the propagation delay is relative to the frame transmission time for the short range system. The disruptions in some of the propagation lines, illustrated by a rotated 'Z' symbol, indicate that the packet has received at least one error in transmission.

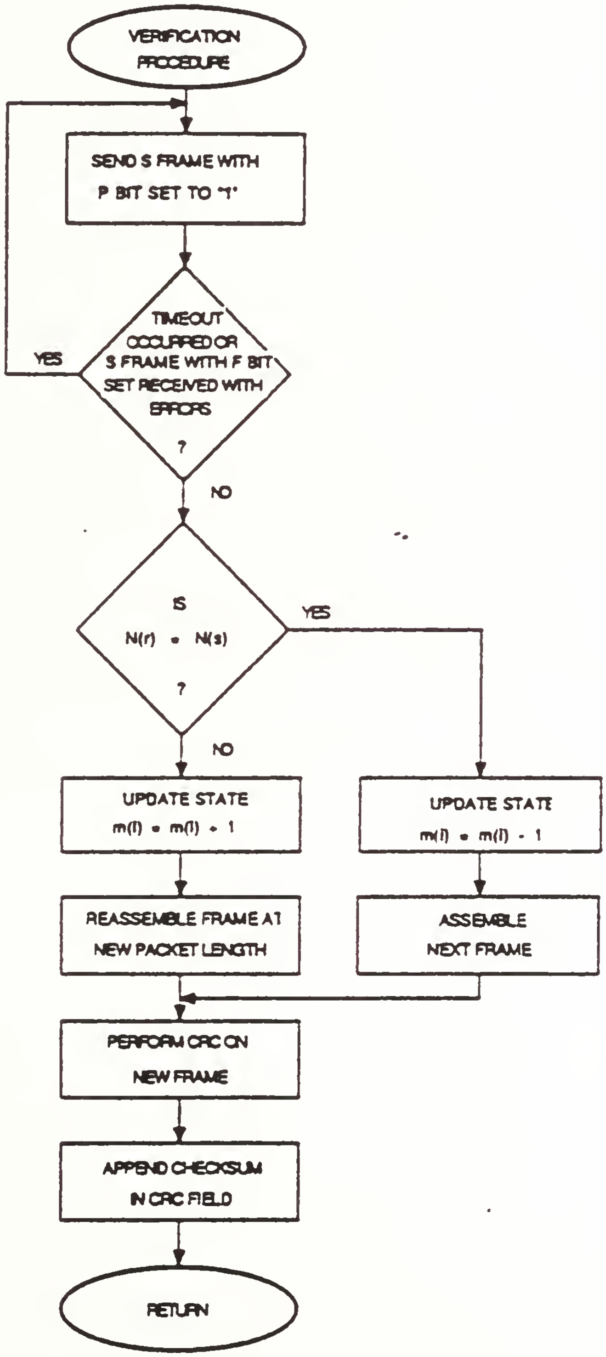

Figure 5. Verification procedure and packet length modification

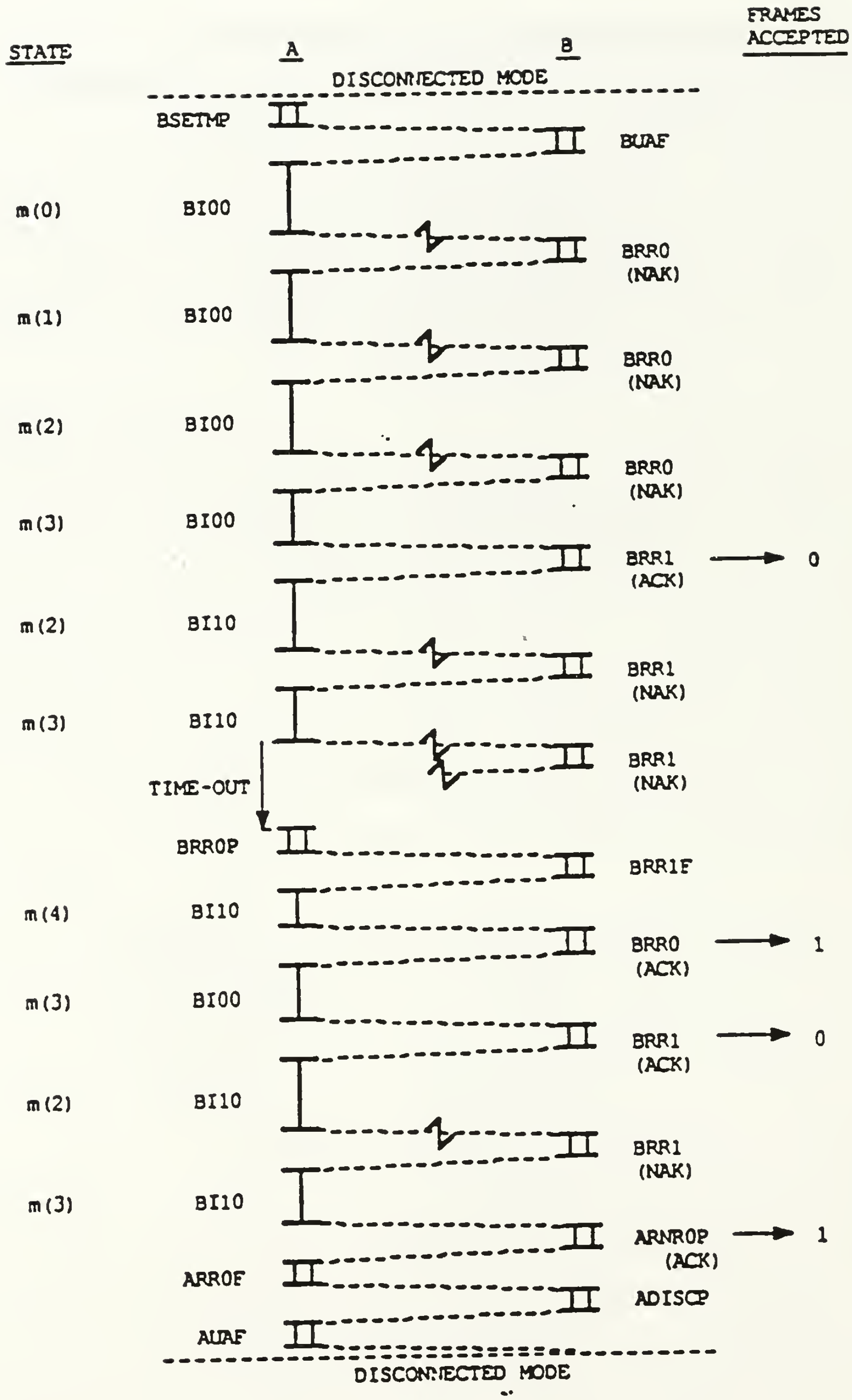

Figure 6 Adaptive SW - an example of one-way information transfer

Often a quick exchange of S or U frames is desired for a station to verify that the frame it is attempting to send is the same as the frame that the remote station is expecting to receive. This exchange can best be handled by using the P/F bit of the control field which initiates the check-pointing mechanism. To force the remote station to respond with either an S or a U frame, the I frames are restricted to only containing commands. This restriction never allows an I frame to be used in responding to a received frame with the P bit set to '1'.

The notations used in the example of protocol operations to describe the frames are the same notations used by Carlson [2]. The protocol established for stations A and B in figure 6 is a multi-level adaptive SW, with each station using a four-level structure specified in table 1. The adaptive structure is implemented by use of the state vector specified in row three of table 1.

Adaptive structure levels	$L(0)$	$L(0)$	$L(0)$	$L(1)$	$L(2)$	$L(3)$
Maximum information field length (bits)	1000	1000	1000	500	250	125
State vector	$m(0)$	$m(1)$	$m(1)$	$m(3)$	$m(4)$	$m(5)$

Table 1. Adaptive SW structure representation and state vector

Station A initiates the communication session by sending a set mode command frame with the P bit set to '1'. Station B is ready to receive data and responds with an Unnumbered Acknowledgment (UA) response frame. Using its own address in the frame, B is indicating that this is a response frame, and the F bit is set to '1' in response to the P bit in the command from A. Upon transfer of the UA frame, B sets its $N(s)$ and $N(r)$ values to zero, initializes its current state to $m(0)$, and enters the information transfer state. When A receives the UA response from B, it performs the same initializations and begins transferring I frames.

The stations start transferring frames according to the adaptive SW protocol, which requires they stop after each transmitted frame and wait for a response. Since B does not have any data to transfer to A during this session, B provides a response for each frame received using a supervisory frame and the appropriate $N(r)$ value. Essentially, B provides an ACK for each frame received without errors and NAK for each frame received with errors. For the system errors caused by transmitted frames that are lost and not received, recovery is performed by activating the time-out function.

A transmits the first I frame BI00, at the length specified by the current state $m(0)$. Since this frame is received at B with errors being detected, B responds with a NAK, by setting the $N(r)$ value to the $N(s)$ value of the frame just received. Upon receipt of frame BRRO, A updates its state by moving one state to the right. The current state is now $m(1)$, which specifies that the frame is to be retransmitted using the same packet length constraint. The attempted data transfer continues, and after the third NAK is received by A, an updating of the current state to $m(3)$ specifies a modification to the frame length prior to retransmitting the frame. To modify the length, A divides the information field into two blocks, places the first block into the information field, performs a CRC on the new frame, appends the new checksum, and transmits BI00. The second block of information is now the next block of information waiting to be transferred.

After the successful transfer of BI00 and the ACK response is received from B, A updates its state to $m(2)$. This state specifies that the maximum frame length is increased to the next level. Therefore, A assembles and transmits frame BI10 at the length associated with level $L(0)$. Frame BI10 is received at B with errors and B responds with a NAK. After updating its state to $m(3)$, A must modify the frame length to the length associated with level $L(1)$ in the same manner as before. A then retransmits the modified frame BI10. The frame BI10 again suffers transmission errors, so B responds with a NAK. Since the NAK sent by B is never received at A, the time-out function at A expires and a recovery action is initiated. In this case, A sends a supervisory type command frame BRR0P, with the P bit set to '1'. This ensures the remote station B is still active and updates the $N(r)$ value. B responds with BRR1F, with he F bit set to '1' in response to the set P bit of the previous frame.

Data transfer from A to B continues with the length of the I frames increasing and decreasing as specified by the state vector. After A has successfully transferred frame BI10, station B is interested in ending the communications session. B sends the command ARNR0P with the P bit set, indicating that B is not ready to receive any additional data from A. A responds with ARR0F indicating receipt of the command frame. B then initiates the disconnect procedures with station A. When A responds with the UA response, both stations enter the disconnect mode.

ADAPTIVE SW SIMULATION

The methods available to determine the throughput efficiency of any ARQ protocol, short of using or building an actual packet switching data communications network and collecting empirical data, are through analysis and computing numerical results, or through simulation. The difficulty in performing an analysis of the system operating with an adaptive protocol is in being able to mathematically or statistically model the system properly. Often, to model all states and processes of the system successfully, certain assumptions and preconditions must be made. Many times the assumptions pose no restrictions on the results, and the analysis results closely resemble actual system performance.

Simulation of the adaptive protocol provides another method of determining the throughput efficiency of the system. To simulate the adaptive protocol operating on a system, the system operation and parameters must be reproduced. In addition, the modeling of the statistical nature of the channel allows the simulation to reflect the actual efficiency of the system. As in analysis, assumptions and preconditions are made to perform the simulations, and to the extent that these conditions are accurate, the simulation results may be indicative of actual system performance.

The simulation model was designed to produce results from which the throughput efficiency of the system operating with one of the adaptive SW strategies could be computed. The simulation modeled a point-to-point, RF data communications system with only two stations. The model is for a system operating over a short distance where the propagation delays is very short relative to the time required for packet transmission. The bit rate for the data transfer is 4800 bit/s. The computer systems modeled in the simulation are capable of processing the packets with almost negligible delay, and they are assumed to be dedicated to the data communications network. The RF channel connecting the stations is modeled to be a noisy channel, and the probability of bit error for the channel can be specified. To model the adaptive SW protocol, the packet sizes had to be controllable, and the overhead associated with each packet held constant. The conditions used for the system in developing the simulation model are

- Only one station transfers data.
- The station transmitting data is always saturated with information.
- The ACKs and NAKs are only 48 bits, and are always received without error.
- Delays associated with the processing time are considered negligible and are not used for the throughput efficiency calculations.
- The bit errors which occur during transmission are considered independent.

These conditions were applied to the model to provide results which could be used to calculate the throughput efficiency.

Three adaptive strategies were used in the simulation, and all three showed a higher throughput efficiency than the non-adaptive SW when the P_b was greater than 10^{-3}. The three adaptive strategies were different, and the relative merit of each can be seen from the results. The results indicate that all the adaptive strategies are very similar in throughput efficiency when the channel P_b is less than about 10^{-3}. For data transfer operations when P_b is larger than 10^{-3}, the multi-level strategy with the state vector that was a single segment representing a structure with the same ascending and descending steps between levels provided the best throughput efficiency. This strategy is referred to as 'multi-level strategy 1' in figure 7. The multi-level strategy with the state vector composed of multiple segments, representing a structure with different ascending steps than descending steps between levels, indicated a throughput value consistently less than or equal to the multi-level strategy with a single segment state vector. This strategy is referred to as 'multi-level strategy 2' in figure 7. The two-level strategy indicated the worst performance of the adaptive strategies, but as the P_b became very high, approaching 10^{-2}, the efficiency curve appeared to level off. This leveling off might indicate that the two level strategy might be quite effective for large swings of the channel P_b.

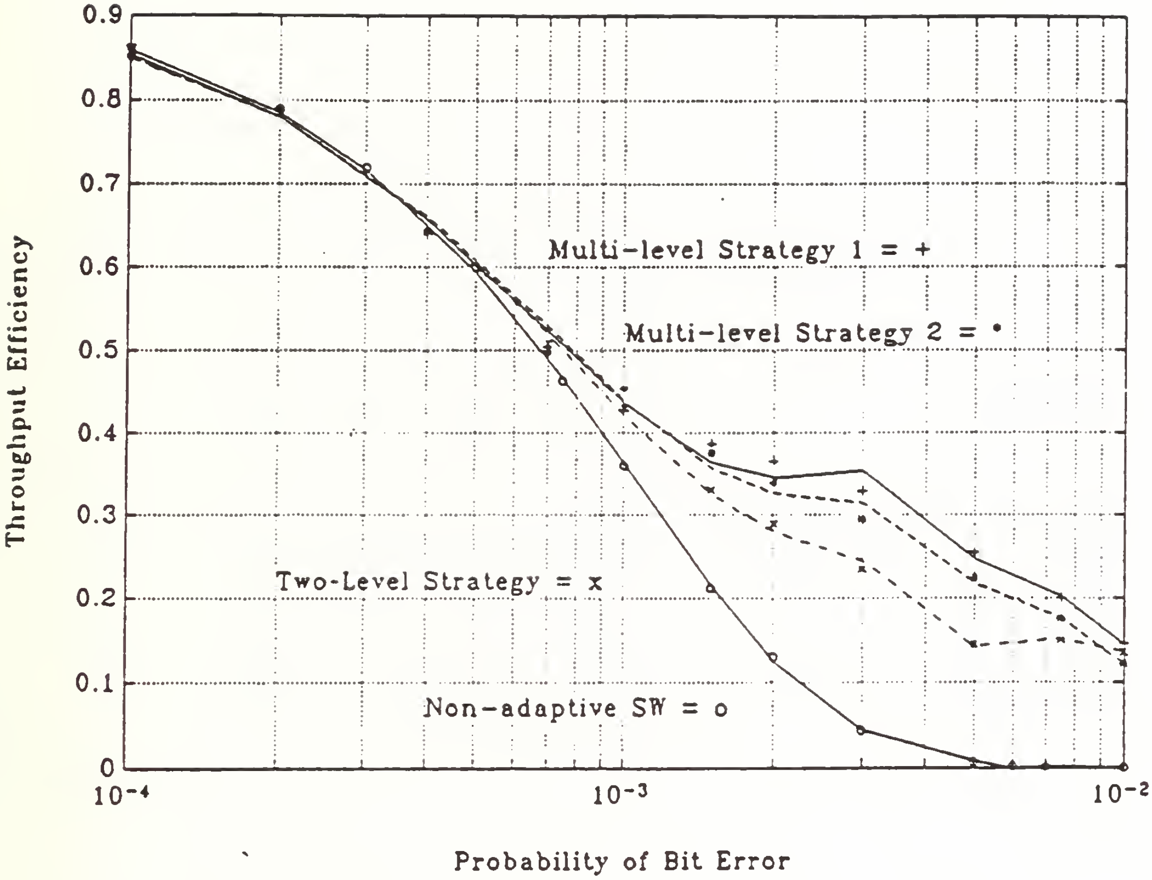

Figure 7 Simulation throughput efficiency vs probability of bit error P_b

THE NETWORK LAYER

The network layer of the OSI model, in conjunction with the data link layer, is responsible for providing the upper layer a virtual link over which messages may transferred between stations. The network layer is responsible for properly routing the packets from sender to receiver, and performs a switching functions to determine where to send all the packets it receives Packets outbound from the station are sent out over the appropriate link via the data link layer, while inbound packets are relayed to the next higher layer, the transport layer Along with this function, the network layer performs flow and congestion control by determining when to accept packets from the higher layers and transmit them on to other stations in the network. It is this layer that is the most conceptually complex.

As mentioned earlier, the mobile nature of the maritime communications stations will pose several problems. Not only will the network need to adapt to the changing availability of hosts and routers on a time basis, but must adapt to the changing geographic configuration of the network due to the movement of ships across the seas. Addressing plans must allow unambiguous identification of any ship-borne host computer on a global basis while routing packets internationally throughout the changing network topology to the proper recipients.

It is desirable to keep the Network Service Access Point (NSAP) address identifying each mobile system constant Stations desiring to communicate with a mobile system need only address their messages to the appropriate NSAP Although each mobile host computer is uniquely identified by its NSAP, the subnetwork point of attachment (SNPA) for the maritime system will change as the ship moves into or out of coverage provided by overhead satellites or other ships [7] As such, the network layer must provide a dynamic means to determine what external routers are reachable by the system based on cost, quality of service, transit delay and security constraints in order to determine the next-hop or destination router for the packet being transferred.

The Aeronautical Telecommunications Network (ATN) uses a distributed-adaptive procedure that may be suited for use by the maritime services. Each router within the ATN is capable of determining the network topology without reliance on a central routing data base. Each router maintains a routing information base that is dynamically maintained through the ATN IS-IS and ES-IS routing information exchange protocols. This routing information base is reduced to a forwarding information base which contains a set of forwarding paths to each known location based on cost, quality of service, policy and other constraints [8]

Further study is required to determine the relative merits of the dynamic routing protocols and routing domain topology options for interconnecting the components of the mobile maritime communications network.

CONCLUSIONS

In mobile communications environments, the system typically operate in high noise environments and are affected by signal fading and interference. The system may operate in the bit error rang of 10^{-3} to 10^{-2} for large periods of time. Automatic repeat request (ARQ) protocols are often used at the data link layer to provide error-free communications links between stations under varying link conditions. The information throughput efficiency is highly link dependent; as the noise in the channel increases, the throughput decreases significantly. It has been shown that the addition of an adaptive feature that increases or decreases the amount of information transmitted within each frame can provide acceptable throughput rates under high channel noise conditions, as compared to the non-adaptive strategy. Because this adaptive feature may be added as a software upgrade to the existing systems, the system designer is allowed the flexibility to further refine the throughput efficiency for his system and the link conditions to which it is subjected.

The network layer must provide means to reliably route packets internationally as the topology of the network changes in time and geographically. Considerable research has been devoted the internetworking of host computers and routing topologies within the Aeronautical Telecommunications Network (ATN). Much of this may have a beneficial impact on the development of routing protocols in the maritime services. Further research is required to determine the relative merits of the routing domain options and routing protocols used within ATN and how they apply to the maritime services.

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